encoding type, packet loading, IP and RPT ports, etc. The default voip profile (voip { 0 0 }) value applies only when a specific parameter value is not specified by an IPDC message.

Routing is controlled from the signaling gateway (for example, Lucent Softswitch), which issues instructions for establishing the TDM/IP/RTP connection between the TAOS units involved in the call. Once this connection is established, the TAOS units pass RTP packets, bidirectionally, across the packet network.

Configuring IP routing for IPDC call processing

There are two methods for implementing IP routing for IPDC VoIP call processing:

- Using RTP listen IP addresses selected by the TAOS unit
- Using RTP listen IP addresses selected by the signaling gateway

When the TAOS unit selects its own listen IP address using RMCP/AMCP messages (see "Using RMCP/AMCP messages to route VoIP calls" on page 2-39), VoIP packet routing and configuration is managed in the same way as H.323 VoIP call processing. See "Packet routing for H.323 VoIP calls" on page 2-29 for details.

When the signaling gateway is configured to perform equal-cost routing across multiple IP addresses (each associated with the Ethernet slot cards in the TAOS unit), each Ethernet port must be assigned IP addresses residing on different logical subnets. In this instance, the signaling gateway determines what IP address to use for the call, with based weighting algorithms used by the media gateway controller application.

Packet routing for IPDC VoIP calls controlled from the signaling gateway

To load balance IPDC VoIP calls across Ethernet slot cards (have the signaling gateway allocate IP addresses in an equal fashion) all Ethernet slot card IP addresses must reside on a different logical subnet within the same TAOS unit. Since routing is controlled by the media gateway controller application on the signaling gateway, this is the only configuration required on the TAOS unit. For this to work, the following rules apply:

- The value of the system-ip-addr parameter for each TAOS unit must:
  - Be unique
  - Be on a logically different subnet from any IP address assigned to any Ethernet card in any MultiVoice network
- No two Ethernet slot card IP addresses assigned to the same TAOS unit may reside on the same logical subnet
- The TAOS unit's IP address must be assigned to SS7-signal packets

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The Ethernet IP address configuration, illustrated by Table 2-16, is an example of how IP addresses should be assigned to Ethernet cards to allow equal-cost routing to be done by the signaling gateway.

Table 2-16. Ethernet IP address table for TAOS unit-173

System IP address	Ethernet interfaces	IP address
192.168.35.173	{{0 0 0}0}*	192.168.35.173/24
	{{1 3 1}0}	208.168.25.173/24
	{{1 4 1}0}	208.168.15.173/24
	{{1 5 1}0}	208.168.5.173/24

* This is the soft IP address assigned to the TAOS unit.

When VoIP data is passed across the packet network between two TAOS units, the source address contained in the SS7 signaling transport packets is used to establish the return path for VoIP call data sent back to the originating TAOS unit. This source address is the IP address where intermediate network routers send data in response to SS7 VoIP data transmissions. To achieve equal-cost routing for IPDC VoIP calls, that source address must be the IP address (that is, the value assigned to the `system-ip-addr` parameter) of the originating TAOS unit.

For example, the following commands set the system address to the address of a port on an Ethernet card in slot 12:

```
admin> get ip-interface { { 1 12 1 } 0} ip-address
[in IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 }:ip-address]
ip-address = 1.1.1.1/24
admin> read ip-global
IP-GLOBAL read
admin> set system-ip-addr = 1.1.1.1
admin> write
IP-GLOBAL written
```

In addition, you must make sure that VoIP calls can always find a route to the next-hop MultiVoice Gateway on the path to the destination MultiVoice Gateway. The route can be learned dynamically or configured as a static route. Many sites choose to configure default routes for VoIP traffic, so that RTP packets are never dropped because of lack of routing information. For example, the following commands configure a default route named VoIP to a next-hop MultiVoice Gateway at 2.2.2.2:

```
admin> new ip-route voip
IP-ROUTE/voip read
admin> set gateway = 2.2.2.2/24
admin> write
IP-ROUTE/VoIP written
```

The IP address of the TAOS unit is assigned to the SS7 signal packets by setting the value of the `use-system-ip-address-as-source` parameter, in the **`ss7-system`** profile, to yes. This inserts the IP address of the originating TAOS unit in the SS7 signaling

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transport packets before transmission across the packet network. See "Using the ss7-gateway profile" on page 2-46.

Using RMCP/AMCP messages to route VoIP calls

The request modify call parameters (RMCP) and accept confirm call parameters (AMCP) messages are used to modify parameters for RCCP/ACCP controlled (VoIP) calls. The messages can be used to modify the following parameters:

- VoIP encoding type (G.711, G.729, and so forth) with Tag 0x70. Note that TAOS also supports G.723 (5.4 Kbps) encoding for SS7 VOIP calls. Following are the supported values for VOIP encoding:

Encoding type	Value
G.711 μ-law	0x00
G.723	0x04
G.711 a-law	0x08
G.729	0x12

- Packet Loading (frames/packet) with Tag 0x73. Values depend on VoIP encoding type.
- Destination Port Type with Tag 0x65. Note that for IPDC 0.12 VoIP calls, the only supported values for Source (0x65) and Dest Port (0x66) Type tags are SCN (0x00) and RTP (0x01) respectively.
- Listen IP address with Tag 0x5D.
- Listen RTP port with Tag 0x5E.
- Send IP address with Tag 0x5F.
- Send RTP port with Tag 0x60.

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The table below shows the tags supported for the RMCP message:

Tag	Parameter Description	R / O status
0x65	Source port type (PSTN only)	Required
0x07	Source module number	Required
0x0D	Source line number	Required
0x15	Source channel number	Required
0x66	Destination port type (RTP only)	Required
0x70	VoIP encoding type (new G.723 value supported)	Optional
0x73	Packet loading (value depends on VOIP encoding type)	Optional
0x5D	Destination listen IP address (see Note below)	Optional
0x5E	Destination listen RTP port number (see Note below)	Optional
0x5F	Destination send IP address (see Note below)	Optional
0x60	Destination send RTP port number (see Note below)	Optional

Note The last four tags in the table are required if values are nonzero. In addition, if an IP address tag is present, the matching port tag must also be present.

This requirement also applies to the same tags in ACMP messages listed in the table below. RCCP and ACCP messages have been modified to use the same requirements regarding these tags.

The table below shows the tags supported for the AMCP message:

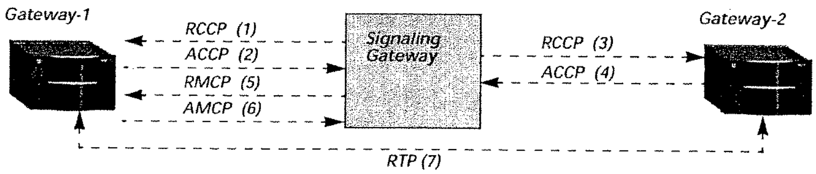
Tag	Parameter Description	R / O status
0x65	Source port type (PSTN only)	Required
0x07	Source module number	Required
0x0D	Source line number	Required
0x15	Source channel number	Required
0x66	Destination port type (RTP only)	Required
0x70	VOIP encoding type (new G.723 value supported)	Required
0x73	Packet loading (value depends on VOIP encoding type)	Required
0x5D	Destination listen IP address	Optional
0x5E	Destination listen RTP port number	Optional
0x5F	Destination send IP address	Optional
0x60	Destination send RTP port number	Optional

Tags 0x70 and 0x73 are required in ACMP messages because RMCP also queries the information for a VoIP call.

Send IP address and Send RTP port tags

With RMCP support of Tags 0x5F and 0x60, the TAOS unit can allocate its own listen IP addresses and RTP ports. The exchanges used in this process are shown in Figure 2-1:

Figure 2-1. Example IPDC message exchanges



IPDC messages to establish RTP listen addresses and ports are exchanged as follows:

1. The signaling gateway sends an RCCP message to Gateway-1, in which the RTP port [n] is either not specified or is 0, but with no IP address or RTP port tags.
2. Gateway-1 returns its RTP listen IP address and RTP port to the signaling gateway in an ACCP message, using tags 0x5D and 0x5E.
3. The signaling gateway sends an RCCP message to Gateway-2, in which the Destination listen IP address and Destination listen RTP port number obtained from Gateway-1 are specified.
4. Gateway-2 returns its RTP listen IP address and RTP port to the signaling gateway in an ACCP message, using tags 0x5D and 0x5E.
5. The signaling gateway sends an RMCP message to Gateway-1, in which the Destination listen IP address and Destination listen RTP port number obtained from Gateway-2 are specified.
6. Gateway-1 returns an AMCP message to the signaling gateway.
7. RTP communication commences between the Gateway-1 and Gateway-2.

Related routing issues

For all VoIP calls, it is important to avoid routing RTP traffic through the TAOS unit's shelf- controller. For that reason, when allowing the TAOS unit to allocate its own RTP address, you must set the `system-ip-addr` parameter in the `ip-global` profile to an interface address other than the default zero address (which defaults to the shelf-controller Ethernet port). For example, the following commands set the system address to the address of a port on an Ethernet card in slot 12:

```
admin> get ip-interface { { 1 12 1 } 0} ip-address
[in IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 }:ip-address]
ip-address = 1.1.1.1/24
admin> read ip-global
IP-GLOBAL read
admin> set system-ip-addr = 1.1.1.1
admin> write
IP-GLOBAL written
```

In addition, it is important that VoIP calls can always find a route to the next-hop MultiVoice Gateway on the path to the destination MultiVoice Gateway. The route

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can be learned dynamically or configured as a static route. Many sites choose to configure default routes for VoIP traffic, so RTP packets will never be dropped due to lack of routing information. For example, the following commands configure a default route named voip to a next-hop gateway at 2.2.2.2:

```
admin> new ip-route voip
IP-ROUTE/voip read
admin> set gateway = 2.2.2.2/24
admin> write
IP-ROUTE/voip written
```

Reporting IPDC VoIP call statistics

A TAOS unit operating as a network access server (NAS) can report VoIP call statistics in the output of the NAS messaging interface. IPDC VoIP call statistics are reported once a call is cleared. The source that originates call clearing can be either the signaling gateway or the TAOS unit.

Supported tags for reporting statistics

IPDC 0.12 statistics tags are reported when the signaling gateway or a TAOS unit clears calls under the following conditions:

- The access server initiates a call teardown using an RCR message
- The access server acknowledges a call teardown using an ACR message for packet-based calls

Table 2-17 shows statistic-related tags from IPCD 0.12 that are currently supported by the MultiVoice Gateway with their descriptions:

Table 2-17. Supported Statistics Tags (IPDC 0.12) (Page 1 of 2)

Tag	Description
0x91	Number of Real-Time Protocol (RTP) audio packets sent and received by the APX.
0x92	Number of RTP audio packets that failed to reach the APX, determined by missed sequence numbers.
0x93	Number of audio bytes in the RTP payload sent by the APX.
0x94	Number of audio bytes received in the RTP payload that failed to reach the APX. Because the number of bytes per packet is variable, this value can only be estimated based upon an average packet size multiplied by the number of non-received packets. The control server can estimate this value with the information supplied.
0x9D	Number of RTP audio packets received.
0x9E	Number of audio bytes received in the RTP payload.

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Table 2-17. Supported Statistics Tags (IPDC 0.12) (Page 2 of 2)

Tag	Description
0xA3	Estimated interarrival jitter (in milliseconds), which is computed as follows: `J = J + (D - J)/16` where `D =	R (i) - R(i-1) - T |`, `R (i)` is the arrival time of the received packet `i`, and `T` is the theoretical difference of departure time between two consecutive packets at the source. For example, `T` is 5 ms for G711 1 frame per packet, `T` is 10 ms for G729 1 frame per packet, and `T` is 40 ms for G729 4 frames per packet.

Unsupported tags for reporting statistics

Table 2-18 shows statistics-related tags from IPDC 0.12 that the MultiVoice Gateway does not support:

Table 2-18. Unsupported Statistics Tags (IPDC 0.12)

Tag	Description
0x95	Number of signaling packets sent and received.
0x96	Number of signaling packets dropped.
0x97	Number of signaling bytes sent and received.
0x98	Number of signaling bytes dropped.
0x99	Estimated average latency.
0x9F	Number of signaling packets received.
0xA0	Number of signaling bytes received.

Call statistics reporting

IPDC 0.15 specifies that the statistics tags are optional, and they are reported in the following cases of call clearing.

- The packet statistics information should be included when the access server initiates a call teardown via RCR message.
- The packet statistics should be included for packet-based calls when the access server acknowledges a call teardown via ACR message.

The TAOS unit reports the statistics in the above two cases when the statistics are available.

ss7nmi debug-level command

The TAOS unit reports VoIP call statistics in the output of the ss7nmi debug-level command. When the command is entered with the -s option, the results displayed include the number of release channel request (RCR) and release channel completed (ACR) messages sent with and without VoIP call statistics, and the number of unknown SS7 VoIP messages. In the following example, new statistics reported for IPDC VoIP calls are shown in bold type:

```
admin> ss7nmi -s
SS7 NAS Messaging Interface (NMI) statistics:
        Initialized successfully:                   Yes
        Total number of internal errors:            0
        Level of diagnostics:                       0
Signaling Layer:
        Current link state:                         UP
        Last generated transaction ID:              182
        Timer T305 (RST1):                          1000 ticks - idle
        Number of protocol version errors:          0
        Number of 'message reject' received:        0
        Number of bad packets received:             0
        Number of unknown messages:                 0
        Number of unknown SS7Voip messages:         0
        Number of resource conflicts:               0
        Number of release race conditions:          0
        Number of RCR with stats sent:              0
        Number of RCR without stats sent:           0
        Number of ACR with stats sent:              36076
        Number of ACR without stats sent:           0
Data Transport Layer:
        Number of link fail-overs:                  0
        Number of persistent errors:                0
        Last error:                                 No Error
        Last error timestamp:                       [09/02/1999 00:00:00]
```

Verifying IP route configuration

To verify the IP route configuration, check the Ethernet port caches and IP caches for each IP interface defined for an TAOS unit.

Verifying VoIP port caches

After creating routes for VoIP packet processing, run the Ipportmap command on the TAOS unit to verify the routing between a port on the Ethernet card and a destination TAOS unit. This command must be entered while calls are in progress, as in the following example:

1 Open a session with an Ethernet card on the TAOS unit. For example, if the Ethernet card is in Shelf 1, slot 2 of your TAOS unit:

```
admin> open 1 2
ether-1/2>
```

2 With a call in progress, enter the ipportmap command to verify port mappings:

```
ether-1/2> ipportmap -m
Port  Proto  Addr              Shelf/Slot       Refcnt
1469  UDP    192.168.35.131/32  1/6/0/0         12503
```

When the ports are mapped properly, the output displays the following information in these fields:

Field	Output
Addr	This is the fully qualified destination IP address of the TAOS unit that is using the network port on this Ethernet card to route VoIP packets.
Shelf/Slot	This is the shelf and slot address of the DSP card on the local TAOS unit which processes VoIP packets sent and received across the network.
Refcnt	This is the total number of packets received from the distant TAOS unit. The Refcnt field continues to increment while the call is in progress.

Verifying VoIP route caches

Enter the ipcache command to verify internal packet routing between an Ethernet card and a MultiDSP slot card on the TAOS unit. This command must be entered while calls are in progress, as in the following example.

1. Open a session with a the MultiDSP slot card. For example, if the DSP card is in shelf 1, slot 6 of your TAOS unit:

```
admin> open 1 6
madd-1/6>
```

2. With a call in progress, enter the Ipcache command to verify that the proper IP route cache was created:

```
madd-1/6> ipcache cache
Hash Address        Gateway         Shelf/Slot Type    MTU  Switched
142  192.168.35.173 208.168.15.173    1/2      STATIC  1500 512
     Forward Stats: To Slots 256, To Shelf 1
     Mem Usage: Allocated 1k bytes
                Free block count 24
```

When packet routes have been properly cached on the DSP slot card, the output display the following information in these fields:

Field	Output
Address	This is the IP address of the far-end TAOS unit that is sending VoIP packets to the local TAOS unit.
Gateway	This is the IP address of the network router, or the Ethernet card on the far-end TAOS unit, which is used to establish the packet network connection to the far-end TAOS unit.

Field	Output
Shelf/Slot	This is the shelf and slot address of the Ethernet card on the local TAOS unit which is used to establish the packet network connection with the far-end TAOS unit.

Trunk configuration

Trunk configuration controls how network signals are processed by the TAOS unit for VoIP calls. For calls originating from SS7 networks, the TAOS unit is configured to let the signaling gateway (manage network signal processing. For calls processed using H.323, the TAOS unit is configured to detect and respond to call progress signals from a PBX/PSTN.

Interoperating with an SS7 signaling gateway using IPDC

For every TAOS unit that interoperates with a signaling gateway, you must configure the following:

- IP interface to the SS7 signaling gateway running IPDC.
- IPDC messaging interface (SS7 profile).
- T1 line settings for SS7.

For more details on configuring the SS7 interface see the *APX 8000/MAX TNT Physical Interface Configuration Guide.*

Using the ss7-gateway profile

The signaling gateway and TAOS unit communicate over a TCP/IP link. The messaging interface can be a single or dual TCP connection between the TAOS unit and the signaling gateway. When the messaging interface initializes, it opens TCP connections to the specified addresses and ports of the signaling gateway. The TAOS unit keeps the TCP connections open as long as the unit is up and the IPDC messaging interface is enabled. Following are the parameters (shown with default settings) for configuring the messaging interface:

```
[in SS7-GATEWAY]
enabled = no
control-protocol = ipdc-0.X
primary-ip-address = 0.0.0.0
primary-tcp-port = 0
secondary-ip-address = 0.0.0.0
secondary-tcp-port = 0
bay-id =
system-type = IASCTNT1B
transport-options = { 0 1000 3000 30000 7 6 }
use-system-ip-address-as-source = yes
```

Parameter	Setting
enabled	Whether the interface is enabled or disabled the interface. When set to no (the default), the interface is disabled. When set to yes, the interface is enabled if the primary-ip-address and primary-tcp-port also have valid values. Changing the setting from yes to no closes the signaling links but does not disconnect active SS7 calls.
control-protocol	The interface control protocol used for communications between the TAOS unit and the signaling gateway. The specified protocol provides call control for setting up, tearing down, and managing calls between the PSTN and the TAOS unit. The TAOS unit must be licensed for the appropriate code for the required control protocol to communicate with the signaling gateway. For VoIP, the TAOS unit must be licensed for IPDC 0.12, and the value of Control-Protocol should be set to ipdc-0.X.
primary-ip-address primary-tcp-port	IP address and TCP port to use as the primary IPDC interface. These settings are required to enable the messaging interface.
secondary-ip-address secondary-tcp-port	IP address and TCP port to use as the secondary IPDC interface. Typically, the primary and secondary address and port configurations point to the two Ethernet interfaces of the signaling gateway.
bay-id	This is an alphanumeric string which identifies the TAOS unit to the media gateway controller application. The content of this field is sent by TAOS unit to the signaling gateway during the device registration process. This parameter is not used with IPDC.
system-type	A device identifier used by the signaling gateway to identify the TAOS unit. This parameter is used for registration purposes only. The content of this field must be recorded on the signaling gateway.
transport-options	This subprofile tunes SS7 L2 timers.
use-system-ip-address-as-source	This parameter assigns either the TAOS unit IP address or signaling gateway address as the source address for SS7 signaling transport packets. The value identifies the IP address of the destination where intermediate network routers should direct data in response to SS7 VoIP/data transmissions. When this parameter is set to yes, the default, SS7 signaling transport packets are assigned the TAOS unit's IP address as their source, or outgoing physical interface address. When this parameter is set to no, SS7 signaling transport packets are assigned the signaling gateway IP address.

Transport-options subprofile

The transport-options subprofile allows users to make occasional changes in the operation of SS7 L2 (level 2) timers. The level 2 portion of the message transfer part

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(MTP Level 2) provides link-layer functionality. It ensures that the two end points of a signaling link can reliably exchange signaling messages. It incorporates such capabilities as error checking, flow control, and sequence checking.

These timers manage the wait/response intervals for these various singling link processes. Changing these values are useful when customers need to use nonstandard values during system integration and for fine-tuning of their network. This subprofile contains the following fields:

```
[in SS7-GATEWAY:transport-options]
device-id = 0
t1-duration = 1000
t2-duration = 3000
t3-duration = 30000
windows-size = 7
ack-threshold = 6
```

Parameter	Setting
device-id	The logical SS7 command control device where these values apply. This is currently not used.
t1-duration	The value of the ACK delay timer in milliseconds. This timer specifies the maximum delay for an acknowledgement when an I-frame is received. Default value is 1000 (1 second). The value must be less than T2 on the peer. Valid values range from 0-2147483647.
t2-duration	The value of the transmission time-out timer in milliseconds. This timer specifies how long this end point should wait for an acknowledgement. Default value is 3000 (3 seconds). The value must be greater than T1 on the peer. Valid values range from 0 - 2147483647.
t3-duration	Value of the persistent error timer in milliseconds. This timer specifies the maximum duration of attempts to reestablish a link before the transport layer flushes the data queues and sends an error indication up. Default value is 30000 (30 seconds). Valid values range from 0-2147483647.
window-size	The maximum number of sequentially numbered data packets that can be sent while pending acknowledgement at any given time. Default value is 7. Valid values range from 1-63.
ack-threshold	The threshold for triggering an acknowledgment while receiving data packets. As soon as the specified number of packets is received, an ACK is sent back regardless of the value set for the T1 timer. The value of this parameter may not be greater than the window size. Default value is 6. Valid values range from 1-63.

Configuring an IP interface to the signaling gateway

For information about configuring LAN and WAN IP interfaces, see the *APX 8000/ MAX TNT WAN, Routing, and Tunneling Configuration Guide.* That guide also describes standard methods you can use to isolate the interface, such as making the route

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private or applying a route filter to the interface, to be certain that only the SS7 messages cross the link between the TAOS unit and the signaling gateway.

Configuring T1 or E1 lines as SS7 data trunks

To configure T1/E1 lines for SS7, you must set the following parameters, shown with sample settings:

```
[in T1/{ shelf-1 slot-1 7 }:line-interface]
signaling-mode = ss7-data-trunk
incoming-call-handling = internal-processing
[in E1/{ shelf-1 slot-10 1 }:line-interface]
signaling-mode = ss7-data-trunk
incoming-call-handling = internal-processing
```

Parameter	Setting for SS7 data trunks
signaling-mode	Must be set to ss7-data-trunk. A line configured as an SS7 data trunk carries no signaling, so it provides 24 (T1) or 32 (E1) 64-kbps channels. When you specify ss7-data-trunk signaling, the line is registered with the IPDC and the IPDC takes control of the line, telling the TAOS unit when to bring calls up or down.
incoming-call-handling	Must be set to internal-processing. Specifies how the TAOS unit processes incoming calls on this line. This value is the same for both H.323 and IPDC VoIP call processing.

Example of configuring a T3 profile

To configure lines of a T3 card as SS7 data trunks, you must first configure the T3 profile as in the following example:

```
admin> read t3 {1 1 1}
T3/{ shelf-1 slot-1 0 } read
admin> set enabled = yes
admin> set frame-type = m13
admin> set line-length = 0-225
admin> write
T3/{ shelf-1 slot-1 1 } written
```

After configuring the T3 line, configure the individual T1 lines that constitute the T3 line as explained in the next section.

Example of configuring a T1 data trunk

The following commands configure a T1 line as an SS7 data trunk, enabling IPDC to control the line:

```
admin> read t1 {1 1 7}
T1/{ shelf-1 slot-1 7 } read
admin> set line-interface enabled = yes
admin> set line-interface signaling-mode = ss7-data-trunk
admin> set line-interface incoming-call-handling = internal-processing
admin> write
T1/{ shelf-1 slot-1 7 } written
```


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Example of configuring an E1 data trunk

The following commands configure an E1 line as an SS7 data trunk, enabling IPDC, from the signaling gateway, to control the line:

```
admin> read e1 {1 10 1}
E1/{ shelf-1 slot-10 1 } read
admin> set line-interface enabled = yes
admin> set line-interface signaling-mode = ss7-data-trunk
admin> set line-interface incoming-call-handling = internal-processing
admin> write
E1/{ shelf-1 slot-10 1 } written
```

Configuring PRI Tunneling in IPDC (IPDC 0.15)

PRI tunneling allows ISDN layer 3 signaling to be tunneled to a signaling gateway. The external signaling gateway controls the T1/E1 PRI lines terminating on a MultiVoice Gateway.

When the TAOS unit acts an access gateway or trunking gateway, the T1/E1 PRI lines must be made visible to an external signaling gateway. The external signaling gateway uses IPDC for call control on these lines.

In this tunneled PRI signaling scheme, a MultiVoice Gateway handles layer 1 and layer 2 of PRI signaling. All layer 3 Q.931 messages on the D-channel are tunneled to the external gateway by means of an IPDC TUNL message. The bearer channels on the PRI lines are controlled by IPDC call setup and teardown messages.

Requirements

To support PRI tunneling, a TAOS unit requires the following:

- IPDC signaling must be enabled on the TAOS unit (that is, `xcom-ss7` must be set to `enabled` in the base profile).
- The IP address and TCP port to use as the IPDC interface to the SS7 signaling gateway must be specified. Typically, the primary and secondary address and port configurations point to the two Ethernet interfaces of the SS7 signaling gateway. Assign the appropriate IP address to the `primary-ip-address` parameter and the appropriate port number to the `primary-tcp-port` parameter in the `ss7-gateway` profile.
- The `signaling-mode` parameter must be set to `tunneled-pri-signaling` in the `line-interface` subprofile of an `t1` or `e1` profile. This value allows an external signaling gateway using IPDC to perform call control on T1/E1 lines terminating on a TAOS unit. The TAOS unit recognizes and responds to IDSN signaling, with local B channels controlled by an external signaling gateway. All layer 3 Q.931 messages are tunneled to the gateway configured in the `ss7-gateway` profile.

Note Only one signaling type can be used on a MultiVoice Gateway or channelized T1/E1 slot card. Only ISDN network terminated (NT) emulation for T1 and T3 lines that are connected to NI-2 and 5ESS/4ESS ISDN switch types are supported.

Reporting PRI tunneling status

The `status` command reports an active tunneled PRI trunk, using the symbol, "`i`". This symbol identifies DS0 connections that use ISDN PRI with layer 3 tunneled signaling to the signaling gateway, as illustrated by the following:

```
0 Connections, 0 Sessions| TNT22 Status
                         | Serial number: 9021340  Version: 9.0a0e0
                         |
                         |  Rx Pkt: 27763
                         |  Tx Pkt: 14688
                         |     col:     2
                         |
                         | 04/06/2000 18:29:42  Up: 0 days, 02:30:20
                         |-----------------------------------------
                         |T-PRI 1/01/01 LA i-------- -------- -------s
                         |                 ^
```

Using tunlpri command options

The tunlpri command is used to report the status of calls processed using tunneled PRI signaling. This command uses the following syntax:

admin> **tunlpri -s**

Using the -s option, the tunlpri command displays module statistics. To enable tunneled PRI diagnostics, use the following diag command to set the desired level for debugging tunneled PRI operations:
diag tunlpri *level*

Following are the values you can specify for *level*:

Debug level	Specifies
0x00	Diagnostic output is disabled. No debugging information is collected.
0x01	Report errors only. Collect only high level error information as errors occur.
0x02	Show basic debugging traces. Collect session logs.
0x04	Dump Tunnel messages. Collect the IPDC TUNL messages sent to and received from the SoftSwitch.
0x08	Show detailed debugging traces. Collect full session logs, including low-level processing information for tunneled PRI signaling.

The following example illustrates the output of the tunlpri command when debug level four (0x04) is specified:
admin>**tunlpri -s**

```
Tunneled PRI Module statistics:
 Interface initialized and ready:        Yes
    Current level of diagnostics:          15
Message count:
 Received from L2  :                     1068
 Sent to L2        :                     754
 Received from Tunl:                     945
 Sent to Tunl      :                     996
```


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```
Errors:
 Errors at startup:                 0
 Warnings:                          0
 Module usage errors:               0
 NULL pointers:                     0
 Control Bus errors:                72
 Buffer pools errors:               0
 Protocol errors:                   0
 Total:                             72
APX6>
```

Modifications to the `ss7nmi` *command*

The `ss7nmi` debug-level command reports TUNL message statistics when entered as follows:

admin> **ss7nmi -m**

When the command is entered with the `-m` option, the results displayed include the number of tunneled PRI (TUNL) messages sent or received by the TAOS unit. The `ss7nmi` debug command includes the following options specifically for IPDC Tunneling debugging:

Options	Specifies
-m	Show TUNL message statistics
-mr	Reset TUNL message statistics
-n	Show active NLCBs (transactions)
-r [address]	Show the status of the SS7 circuit(s). When address is specified, show only status for the selected circuit.
-rc	Toggle, enable or disable, resource backtrace collection. By default, this option is disabled.
-rd [address]	Show detailed status of circuit(s). When address is specified, show only status for the selected circuit.
-s	Show SS7 interface statistics
-sr	Reset SS7 interface statistics

To enable tunneled PRI diagnostics, use the following `diag` command to set the desired level for tracing tunneled PRI messaging:
diag ss7nmi *level*

Debug level	Specifies
0x00	Diagnostic output is disabled. No debugging information is collected.
0x01	Report errors only. Collect only high level error information as errors occur.

Debug level	Specifies
0x02	Show signaling link states. Collect information on SS7 link-state changes.
0x04	Show NLCB/transaction states. Collect information on NCLB statuses and transactions state changes.
0x08	Show signaling semantics. Collect information on signaling types associated with each call.
0x10	Display contents of NMI packets. Collect information from network management information packets.
0x20	Show call control interface details. Collect information on the interface used to set up, monitor and tear down each call.
0x40	Show internal task events. Collect information on the low-level processes used for call control.
0x80	Show memory usage. Collect information on the memory allocated by the TAOS unit to process calls.
0x100	Show resource allocation details. Collect information on how TAOS unit resources are allocated for each call.
0x200	Show tunnel basic errors. Collect only high-level tunneling PRI error information as errors occur.
0x400	Show tunnel basic debug. Collect only high-level tunneling PRI debugging information for calls as they occur.
0x800	Dump Tunnel messages. Collect the IPDC TUNL messages sent to and received form the SoftSwitch.
0x1000	Show detailed debugging traces. Collect full session logs, including low-level processing information for tunneled PRI signaling.

The following example illustrates the output of the `ss7nmi -m` command, reporting the TUNL messaging statistics:

```
admin>ss7nmi -m
     IPDC message processing statistics:
        Message code       Received      Sent
        RCR    (0x0011):     152802         0
        ACR    (0x0012):          0    152802
        RCCP   (0x0013):     152847         0
        ACCP   (0x0014):          0    152847
        RMS    (0x0041):          1         0
        NMS    (0x0042):          0        24
        RLS    (0x0043):         28         0
        NLS    (0x0044):          0        29
        NCS    (0x0046):          0         7
```

```
TUNL   (0x007a):     611460     611480
RTE    (0x007d):        111          0
ARTE   (0x007e):          0        111
NSUP   (0x0081):          0          1
ASUP   (0x0082):          1          0
Data collection was started: [04/08/2002 17:24:01]
```

Configuring trunk signaling for H.323 VoIP networks

For every TAOS unit that operates in a H.323 VoIP network and connects to the PBX and PSTN, you must configure the T1 line settings to detect and respond to call progress signaling for H.323 VoIP calls. Set the following parameters, as shown with sample settings:

```
[in T1/{ shelf-1 slot-1 1 }:line-interface]
signaling-mode = inband
robbed-bit-mode = inc-w-200
default-call-type = voip
collect-incoming-digits = yes
t1-inter-digit-timeout = 6000
[in T1/{ shelf-1 slot-1 2 }:line-interface]
signaling-mode = isdn
default-call-type = digital

[in E1/{ shelf-1 slot-1 1 }:line-interface]
signaling-mode = r1-inband
robbed-bit-mode = inc-w-200
default-call-type = voip
number-complete = 7 digits
caller-id = get-caller-id
e1-inter-digit-timeout = 6000
[in E1/{ shelf-1 slot-1 3 }:line-interface]
signaling-mode = isdn
default-call-type = digital
```

Parameter	Setting
signaling-mode	Type of signal received from the T1 trunk. Set this parameter to: • inband for non-PRI T1 trunks using inband with robbed bit signaling. When using inband signaling (T1, MF R2), audible tones are used to transmit DNIS/ANI across the trunk. • isdn for T1/PRI trunks. When using ISDN signaling, DNIS/ANI are transmitted in the ISDN call setup message. • dtmf-r2-signaling for R2 signaling trunks. When using DTM R2 signaling, a DSP is allocated to detect DTMF tones for inbound and outbound calls (see "Enabling DTMF R2 signaling for E1 lines" on page 2-57 for details). • inband-fgd-in-fgd-out for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received and sent in FGD format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details). • inband-fgd-in-fgc-out for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received in FGD format and sent in FGC format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details). • inband-fgc-in-fgc-out for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received and sent in FGC format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details). • inband-fgc-in-fgd-out for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received in FGC format and sent in FGD format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details).
robbed-bit-mode	Type of robbed bit-signaling received from an inband T1 trunk. Set this parameter to inc-w-400 or inc-w-200 for trunks supporting DNIS/ANI.

Parameter	Setting
`default-call-type`	Default calltype for incoming calls, when using non-PRI T1 trunks (inband with robbed-bit signaling), or when the TAOS unit is configured for single-staged dialing. Set the value of this parameter to `voip` if all calls received are processed as VoIP calls. For T1 configurations that don't provide DNIS (in-band), this method is required to map an incoming call as a VoIP call. If other call types (such as, modem calls) are received over this trunk, set the value to `digital`.
`collect-incoming-digits` (T1) `caller-id` (E1)	Enables/disables collection of the Dialed Number Identification Service (DNIS) string for the destination telephone number and Automatic Number Identification (ANI) string of the calling telephone number for an incoming call. For trunks supporting DNIS/ANI, set the value of this parameter to yes. This parameter is ignored when `signal-mode=isdn`. If two-stage dialing is being used with ISDN signaling, then the two-stage dialing parameter must be set to yes. **Note** If DNIS/ANI is present on the T1, you must set this parameter to yes, otherwise the H.323 signaling layer identifies the DNIS/ANI digits as part of the destination phone number dialed by the user. If DNIS/ANI is not desired, provision the PSTN switch/PBX not to send DNIS/ANI.
`t1-inter-digit-timeout` `e1-inter-digit-timeout`	How long the TAOS unit waits after receiving the last digit before declaring DNIS/ANI collection complete, when using inband signaling (T1, MF R2). The TAOS unit waits until this interval has elapsed to ensure it has received all audible tones used to transmit DNIS/ANI across the trunk. It may be set to a value between 100 and 6000msec. This parameter defaults to 3000msec. (3 seconds) and works with the Number-Complete parameter when time-out processing is implemented (for details, see "Enabling collections of variable length dial strings without EOP" on page 2-61). **Note** The `call-inter-digit-timeout` parameter in the `voip` profile is used to control collection of keypad generated DTMF entered by the caller, when using two-stage dialing. It has no effect on the collection of DNIS/ANI for T1 signaling.
`number-complete`	In the E1 profile, the condition that the MultiVoice Gateway uses to determine the length of the dial string. Up to 15 digits can be collected for R2 dial strings without waiting for end-of-pulse (EOP) signaling. Time-out processing can also be implemented by setting this parameter to `time-out` (see "Enabling collections of variable length dial strings without EOP" on page 2-61 for details).

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Note For robbed-bit interfaces, if the `default-call-type` is `voip` in the `t1` profile, then the bearer capability in the setup message is altered to indicate that this is "packetized voice."

In-bound calls received on all channels of this T1 will be processed by the TAOS unit as VoIP calls.

Enabling DTMF R2 signaling for E1 lines

MultiVoice Gateway can process Dual Tone Multi-Frequency (DTMF) tones over R2 signaling trunks to provide support for processing either country-specific R2 signaling (MFC-R2) or DTMF signaling over trunks that support standard R2 signaling.

MultiVoice Gateways can support DTMF R2 signaling generated by smaller European network switches and PBXs. MultiVoice implements DTMF tone processing using the R2 signaling standard defined by the International Telecommunications Union Telecommunication sector standard (ITU-T) Q.400, *Specifications of Signaling System R2 Definition and Function of Signals -- Forward Line Signals.*

A channelized E1 slot card uses one of the following channelized associated signaling (CAS) types:

- R1
- R2 or any R2 variant
- DTMF-R2

Note Only one signaling type can be used on an TAOS unit channelized E1 slot card.

To support DTMF-R2 detection, MultiVoice requires the following:

- Connection to E1 trunks that are attached to a switch that supports the ITU-T R2 signaling standard.
- The switch must generate and/or relay the high-frequency/low-frequency tone combinations generated by normal touchtone dialing to the MultiVoice Gateway.
- R2 signaling must be enabled on the MultiVoice Gateway. Verify that the R2 signaling parameter is enabled—check the base profile for `r2-signaling-enabled`=yes.

Detection of DTMF R2 signals is enabled from the `e1` profile.

DTMF tone detection

When processing tones for DTMF R2 signaling, the MultiVoice Gateway performs as follows:

- Upon detection of an inbound call, allocate a DSP for detecting DTMF tones, capturing DTMF digits as they are received from the switch.
- Upon receipt of an outbound call (from the packet network), allocates a DSP for generating DTMF tones, sending the first DTMF tone for 70ms, followed by 70ms of silence. This tone-silence sequence is repeated until all digits are sent to the telephone switch.

Note A DSP can successfully generate and detect test tones for E1 SS7 continuity testing without impacting detection and processing of E1-R2 signaling. Specifically, the required frequency for the E1 SS7 continuity check, 1780+-20Hz and 2000+-20Hz with the sending tone level of -12+-1dbm0, as defined in ITU

Telecommunication sector standard (ITU-T) Q.724, *Specifications of Signalling System No. 7 - Telephone user part* (1988), International Telecommunications Union, are detected by the DSP. The DSP differentiates between tones in this same frequency ranges which are used for E1-R2 signaling and the E1 SS7 continuity check.

The following is an example of how to enable DTMF R2 signaling on an APX or MAX TNT E1 line slot card.

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read
admin> set signaling-mode=dtmf-r2-signaling
admin> set collect-incoming-digits=no
admin> set e1-inter-digit-timeout=3000
admin> write
E1/{ 1 1 7 } written
```

Using Signaling-Mode parameter for DTMF R2 signaling on E1 lines

One of the settings for the `signaling-mode` parameter in the `e1` profile enables DTMF R2 signaling detection and processing in the `e1` line profile.Setting the value of the `signaling-mode` parameter to `dtmf-r2-signaling` value enables the TAOS unit to recognize and respond to the DTMF R2 signal set during voice and data calls. Once selected, DTMF R2 detection is enabled with the next VoIP call.

The following dependencies apply when `signaling-mode=dtmf-r2-signaling`:

- `collect-incoming-digits` must be enabled (`collect-incoming-digits=yes`).
- Assigning a lower value (such as 600 to 3000) to the `e1-inter-digit-timeout` parameter improves call setup times. Assigning a higher value (such as 3001 to 6000) improves detection of DTMF.
- DTMF R2 detection is only supported when R2 signal processing is enabled for the TAOS unit. The Base profile should contain the following setting:

  `r2-signaling-enabled=yes`

Enabling and debugging Feature Group D signaling support for T1 lines

MultiVoice supports a subset of the Telecordia requirements for Feature Group D (FGD) signaling for Voice over IP processing, such as passing Automatic Number Identification II (II) information, Calling-Party-Number and Called-Party-Number as MFR1 tones on inc-wink signaled trunks. A MultiVoice Gateway can manage interworking between Access Tandem carriers and traditional toll service carriers for VoIP calls. It also provides basic support for trunk-side access with Equal Access dialing capability, pre subscription, and enhanced signaling options for Automatic Number Identification as specified by Requirement GR-690-CORE, *Exchange Access Interconnection FAS 20-24-0000* (Oct. 1995), Telecordia Systems (formerly Bellcore).

Feature Group D access service with equal access multifrequency signaling is characterized by two-stage outpulsing when connection is made through the access tandem. The first stage provides information to the AT for selection of a carrier and the route to take to that carrier. The second stage provides the carrier with both the calling-party-number (and, optionally, ANI) and the Called-Party Number (address or destination number). Overlap outpulsing is used to transmit this information using multifrequency signaling.

Pass-through of equal access signaling can be enabled for T1 inband trunks from the T1 line profile. When FGD signaling is enabled, MultiVoice Gateways can recognize and process the single-dialed access carrier destination, (such as: 1,2025551212 or 1,1010220,2025551212). To support access carrier billing, a MultiVoice Gateway passes the Calling-Party-Number, ANI information digits and Called-Party-Number. The ANI information digits, a two-digit code, classifies the Calling-Party-Number by tariff type (such as coin, 800 service, or POTs).

MultiVoice also manages interworking when connecting VoIP calls between Access Tandem networks and traditional toll service networks. An egress MultiVoice Gateway can be configured to receive Calling-Party Number, ANI information digits, and Called Party-Number from an Access Tandem switch and connect that call to a switched telephone network that supports Feature Group C (FGC), that is, traditional toll service and switching. FGC includes automatic number identification of the calling party, answerback, and disconnection supervision. FGC service predates the breakup of the Bell System.

T1 profile

Feature Group D licensed software for MultiVoice Gateways adds FGD signaling options to the `signaling-mode` parameter in the `t1` profile to enable inband signal processing of FGD signals, and interworking of MultiVoice networks between Access Tandem carriers and traditional toll-service carriers.

fgd-signaling-enabled parameter

Following installation of TAOS, each MultiVoice Gateway must be loaded with licensed software code to enable processing of Feature Group D signaling. When enabled, `fgd-signaling-enabled` parameter appears in the base profile.

During manufacturing or software upgrade of MultiVoice Gateways, the installation binaries used to install TAOS on the TAOS unit asks if FGD support should be enabled. FGD support can only be enabled or disabled by reinstalling the licensed software on the TAOS unit.

When support for FGD is enabled, the `fgd-signaling-enabled` parameter is added to the base profile, as illustrated:

```
admin> read base
BASE read (read-only)

admin> list
[in BASE]

shelf-number = 1
software-version = 8
software-revision = 0
software-level = ""
manufacturer = dba-ascend-mfg
....
fgd-signaling-enabled = yes
```


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Using the signaling-mode parameter to configure FGD signaling

The `signaling-mode` parameter in the `t1:line-interface` subprofile identifies the type of call signal received from the ingress switched telephone network and the type of call signals passed to the egress switched telephone network. New values may be assigned to the Signaling-Mode which enable processing of FGD inband signaling for connecting Equal Access calls, and support interworking between Access Tandem and traditional toll service networks.

The `signaling-mode` parameter may be assigned the following values to enable FGD signaling support:

Parameter	**Setting**
`inband-fgd-in-fgd-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGD format, and connect VoIP calls to the egress switched telephone network, sending call signaling data in FGD format.
`inband-fgd-in-fgc-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGD format, and connect VoIP calls to the egress switched telephone network sending call signaling data in FGC format.
`inband-fgc-in-fgc-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGC format, and connect VoIP calls to the egress switched telephone network sending call signaling data in FGC format.
`inband-fgc-in-fgd-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGC format, and connect VoIP calls to the egress switched telephone network sending call signaling data in FGD format.

Changes made to the `signaling-mode` parameter take effect with the next VoIP call. The following example illustrates how to enable a MultiVoice Gateway to receive call signaling data in FGD format and send call signaling data to the egress switched telephone network in FGC format signalling using the `signaling-mode` parameter.

```
admin> read t1 { 1 1 1 }
T1/{ 1 1 1 } read
admin> list line
[in T1/{ shelf-1 slot-1 1 }:line-interface]
enabled = yes
frame-type = esf
encoding = b8zs
signaling-mode = inband
.........
ss7-continuity = { loopback single-tone-2010 }
admin> set signaling-mode=inband-fgd-in-fgc-out
admin> write
T1/{ 1 1 1 } written
```

For the `signaling-mode` parameter to include FGD signaling options, the MultiVoice Gateway must have licensed software code for FGD processing.

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Feature Group D signaling timing

To ensure wideranging interoperability with available access tandem switches, MultiVoice uses the middlerange Feature Group D Signaling Timing.

- Wait up to 210 msecs for first wink from the Access Carrier. Requirement GR-690-CORE specifies a range of 140-290 msecs.
- Wait up to 5 seconds to receive the first digit. After sending the first wink on receipt of an off hook, the MultiVoice Gateways waits for 5 seconds before reporting a time-out error if the first digit signal is not received.
- Wait up to 4 seconds for a wink from the Access Carrier. Requirement GR-690-CORE specifies Access Tandem switches wait for up to 4 seconds for this signal.

Debugging Feature Group D signaling

To collect debugging information for Feature Group D inband signal processing on a TAOS unit, enter the following commands:

```
TnT01> open 1 1          (where the T1 slot card is)
TnT01> fgdtoggle         (turn on fgd signalling debugging)
TnT01> debug on
```

The debug command displays information similar to the following for Feature Group D signal processing.

```
----------------------------------------------------------------------
0 Connections, 0 Sessions  | TNTFCD Status
                           | Serial number: 9340872 Version: 9.0a0e0
                           |
                           |  Rx Pkt:    1057139
                           |  Tx Pkt:     163995
                           |     Col:       2244
                           |
                           |
                           | 02/29/2000 17:50:27  Up: 7 days, 00:17:30
                           | "T3 Tru+ 1/03/00 LA la la la la la la la
                           | "T3 Fiv+ 1/03/01 LA F------- ------- -------
                           | "T3 CDX" 1/03/02 LA F-------- -------- -----
                           |                     ^
                           |                     |
                           |                     +-- display (F)for FGD line
                           |
                           |
-----------------------------------
```

Enabling collections of variable length dial strings without EOP

The MultiVoice Gateway collects variable-length dial strings without using end-of-pulse (EOP) signaling. In certain areas outside the continental United States where E1 MFC-R2 signaling is used for switched network operations, the length of E.164 addresses vary. End-of-pulse (EOP) detection is not efficient, since the network may be unable to complete the call as a result of network conditions.

Collection of up to 15-digits

MultiVoice Gateways are compatible for use on European telephone systems that use E.164 addresses that are up to 15 digits long, without waiting for an end-of-pulse signal.

Previously, MultiVoice Gateways could be configured to collect dial strings of up to only 11 digits. For European networks using dial strings that were 12 digits or longer, a MultiVoice Gateway could only be configured to wait for the end-of-pulse signal to confirm it received all the dialed digits.

The number of configurable digits to 15 for the E1 line `number-complete` parameter is set in the `e1:line-interface` sub-profile.

The `number-complete` parameter enables detection and collection of up to 15 digits for inbound dialed telephone numbers on MultiVoice Gateways using E1 trunks supporting inband CMF R2.

The parameter now accepts values from `0-digits` through `15-digits`, `end-of-pulse`, and `time-out` as valid entries.

The following example illustrates how to enable the collection of 15 digit dial strings on a TAOS unit:

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read

admin> set number-complete=15-digits

admin> write
E1/{ 1 1 7 } written
```

The following dependencies apply to this parameter:

- The `number-complete` parameter defaults to `N/A` when the `signaling-mode` parameter is assigned any of the following values:
  - `e1-kuwait-signaling`
  - isdn
  - p7
  - dpnss
  - none

Time-out processing

A MultiVoice Gateway can be configured to time-out, followed by pulse signals, as specified by ITU-T Recommendation Q.442, *Specifications of Signalling (sic) System R2 interregister Signalling (sic), Pulse Transmission of Backward Signals A-3, A-4, A-6 or A-15* (1993), International Telecommunications Union.

Using time-out allows a MultiVoice Gateway to delay processing of a dialed number string, even after receiving the last digit, to allowing the resources on the switched network additional time to become available, before continuing with call processing.

To implement this feature, modify entries for the `number-complete` and `inter-digit-time-out` parameters.

The `number-complete` parameter sets the condition the MultiVoice Gateway uses to determine the length of the dial string. For E1 MFC-R2, the MultiVoice Gateway

continues to collect digits until the on/off pulsing using to transmits the dial string is complete.

The `number-complete` parameter now accepts the following value:

Parameter value	Description
`time-out`	Configures the MultiVoice Gateway to reset the network idle timer after the initial digit is received, and then wait for silence. Once silence is detected, wait the interval specified by the `inter-digit-time-out` parameter for next digit. The MultiVoice Gateway continues to collect digits, while waiting for the network idle timer to expire before continuing with call processing.

The following illustrates how to configure a MultiVoice Gateway to determine the length of a dial string using time-out processing.

```
admin> read e1 { 1 1 1 }
E1/{ 1 1 1 } read

admin> list line
[in E1/{ shelf-1 slot-1 1 }:line-interface]

enabled = yes
frame-type = esf
encoding = b8zs
signaling-mode = inband
.........
ss7-continuity = { loopback single-tone-2010 }

admin> set number-complete=time-out

admin> write
```

E1 MFC-R2 signaling is country specific. The `signaling-mode` parameter, and the `country` parameter in the `system` profile, must be set for the country-appropriate signaling in order for the MultiVoice Gateway to properly detect dialed digits.

In the `e1:line-interface` subprofile, the `inter-digit-time-out` parameter controls how long a MultiVoice Gateway will wait after receiving the last digit of a dial string before declaring DNIS/ANI collection complete. When using inband signaling (T1, MF R2), a TAOS unit waits until this interval has elapsed to ensure it has received all audible tones used to transmit DNIS/ANI across the trunk.

The `inter-digit-time-out` parameter accepts values between 100 and 6000msec. This parameter defaults to 3000msec (3 seconds). For configurations supporting E1 MRC R2 signaling, the `inter-digit-time-out` parameter accepts values between 200 and 6000msec.

The following illustrates how to configure the interdigit timer on a MultiVoice Gateway to wait one second (1000msec) in between dialed digits before continuing with call processing.

```
admin> read e1 { 1 1 1 }
E1/{ 1 1 1 } read
```


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```
admin> list line
[in E1/{ shelf-1 slot-1 1 }:line-interface]
enabled = yes
frame-type = esf
encoding = b8zs
signaling-mode = inband
.........
ss7-continuity = { loopback single-tone-2010 }
admin> set inter-digit-time-out=1000
admin> write
```

E1 MFC-R2 signaling is country specific. The `signaling-mode` parameter in the `e1` profile and the country parameter in the `system` profile must be set for the country-appropriate signaling for the MultiVoice Gateway to properly detect dialed digits.

Processing ANI and DNIS for H.323 VoIP

Regardless of whether a TAOS unit is configured for single stage-dialing or two-stage dialing, the master DSP or slave DSP parses the dual-tone multifrequency (DTMF) tones in the following order, based on how the tones were processed by the PSTN/switch:

CLID*DNIS
DNIS
CLID*DNIS*

When the local TAOS unit connects with the destination TAOS unit involved in a call, the reported DNIS and ANI/CLID, which are included as part of the call setup message(s), are used as follows:

String	Description
ANI/CLID	The Automatic Number Identifier or Calling Number IDentification string of the caller's telephone: • At the destination gateway, this number identifies the origin of the call. Once received by the destination gateway, the number is exported for use by gateway applications, passed back to the calling gateway to confirm call setup, etc. • At the local gateway, this number is collected then forwarded to the destination gateway and the MultiVoice Access Manger, where it may be used for authentication, call reporting, third-party billing applications, etc. • At the destination gateway, this number is passed to the PSTN and used for initiating local switched network services that process CLID (such as Caller ID, call waiting, last number redial, etc.).

String	Description
DNIS	The Dialed Number Identification Service string that identifies the called telephone: • At the destination gateway, this number is the destination telephone number dialed by the MultiVoice Gateway. • At the local gateway, this number is the dial string entered by the caller for the destination telephone number.

Configuring 480 ports for G.711-encoded VoIP-only calls

The MultiDSP 288-port slot card (APX-SL-DSP-3) in an APX 8000 or APX 1000 supports the following modes:

- 288 ports of universal port traffic (that is, simultaneous V.110 and V.92 modem, High-Level Data Link Control (HDLC), Personal Handyphone System (PHS), T.38 Fax, VoIP with G.711, G.729(A) audio codecs). This is the default mode.
- 480 ports of G.711 VoIP-only calls for H.323 and IP device control (IPDC) protocols.

Sites that use the G.711 codec in their networks have the option to configure 480 ports, which lowers the cost of network equipment and operation.

To support 480 ports of G.711 VoIP-only traffic, the following conditions must be met:

- The universal gateway must be an APX 8000 or APX 1000 unit.
- The MultiVoice software license for each universal gateway must be enabled. Additional software licenses or hardware are not required.
- The `frames-per-packet` parameter in the `voip` profile must be set to 4 or greater. The maximum number of frames per packet you can set is 10. If a number lower than 4 is set, the value is automatically reset to 4 and a log message is generated.

Note The value specified for the `frames-per-packet` parameter (4 - 10) must be supported by all MultiDSP slot cards in the chassis. If using the IPDC protocol, the softswitch should use four frames per packet for all calls.

Transparent fax and modem

Regular modem (for example, v.90/v.92) and T.38 fax capability are not supported in the 480 port configuration. However, when configured for 480 ports, the slot card uses two frames per packet for transparent fax and modem.

The slot card can handle only a maximum of 96 transparent fax and modem calls per quadrant. When using priority-based call routing (see "Priority-based call routing" on page 2-25), all G.711 calls are routed to the slot card. Even if other MultiDSP slot cards also handle transparent fax, Lucent recommends allowing only a maximum of 96 transparent calls per quadrant.

Compatibility with other MultiDSP slot cards

When using IPDC and the slot card is configured for 480 ports of G.711 VoIP-only traffic, the MultiDSP 288-port slot card can coexist with 48-port and 96-port MultiDSP universal port slot cards that use other codecs.

However, when using H.323 and the slot card is configured for 480 ports of G.711 VoIP-only traffic, only G.711 is supported by all MultiDSP universal port slot cards.

New value for subtype parameter

When the slot card is activated initially, the system checks for the presence of a madd-slot-config profile.

If the profile exists, the system examines the subtype parameter to determine the number of ports the slot card supports. If the profile does not exist, then the slot card operates with the default 288 universal ports.

The values for the subtype parameter are:

Parameter Value	Description
480-voip-ports	Configures the slot card for 480 ports.
288-univ-ports	Configures the slot card for 288 ports. Default mode.

Configuring the slot card for 480 ports

To configure the MultiDSP 288-port slot card for 480 ports, create a new madd-slot-config profile and specify a value for the subtype parameter.

The following example configures the slot card that resides in shelf 1, slot 8:

```
admin> new madd-slot-config { 1 8 0 }
MADD-SLOT-CONFIG/{ shelf-1 slot-8 0 } read
admin> list
[in MADD-SLOT-CONFIG/{ shelf-1 slot-8 0 } (new)]
slot-address* = { shelf-1 slot-8 0}
subtype = 288-univ-ports

admin> set subtype = 480-voip-ports
admin> wr
```

Resetting the slot card

After making changes to the madd-slot-config profile, verify that there are no active calls being processed by the slot card. Then reset the slot card or reset the entire universal gateway. To reset a slot card located in shelf 1, slot 8, proceed as follows:

1. **Check to see if the slot card is currently processing active calls:**

   ```
   admin> modem -i | grep "1 8"
   Modems allocated/in-use:
     Modem { 1 8 3 } ( Up Assign UP UP ENABLE )
   ```

2. **While waiting for active calls to be discontinued, prevent new calls from being routed into this slot card by disabling the modems:**

   ```
   admin> mdmdisable 1 8
   ```

3. **Keep checking the status of the current calls until all calls are no longer being processed:**

   ```
   admin> modem -i | grep "1 8"
   ```

   You should see the following message before resetting the slot card:

S 1405

```
Modems allocated/in-use:
  Modem { 1 8 3 } ( Up Assign UP UP ENABLE )
```

4 **Bring the slot down with the following command:**

```
admin> slot -d 1 8
```

5 **Remove the leftover profiles from the system with the following command:**

```
admin> slot -r 1 8
Slot 1/8, state change forced
```

6 **Activate the slot card with the following command:**

```
admin> slot -u 1 8
```

7 **To show status of the slot card, enter the following command:**

```
admin> sh
Controller { left-controller } (PRIMARY):
                          Reqd  Oper   Slot Type
{ shelf-1 slot-1 0}       UP    UP     8t1-card
{ shelf-1 slot-2 0 }      DOWN  RESET  ether3-card
{ shelf-1 slot-7 0 }      DOWN  RESET  t3-card
{ shelf-1 slot-8 0 }      UP    UP     madd3-voip-480
```

Reverting to universal port mode

To revert to using 288 universal ports, use the following commands. The example assumes that the slot card has been installed in shelf 1, slot 8:

```
admin> read madd-slot-config { 1 8 0 }
MADD-SLOT-CONFIG/{ shelf-1 slot-8 0 } read

admin> list
[in MADD-SLOT-CONFIG/{ shelf-1 slot-1 0 } (new)]
slot-address* = { shelf-1 slot-8 0}
subtype = 480-voip-ports


admin> set subtype = 288-univ-ports
admin> wr
```

After making changes to the madd-slot-config profile, reset the slot card or reset the entire universal gateway (see "Resetting the slot card" on page 2-66 for details).

In-call DTMF detection for IPDC

You can configure your TAOS unit to allow Softswitch to direct the MultiVoice Gateway to perform in-call DTMF detection and notification while a packet call is in progress. This is accomplished by modification of the RCCP, RMCP, and NTN messages. Any DTMF digits entered during the call while DTMF detection is enabled are still played out to the other party.

Note In-call DTMF detection is supported for packet calls, but not for time-division multiplexing (TDM) calls. Also, this feature requires obtaining a pre-paid billing application.